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الَّذِينَ آمَنُوا وَعَمِلُوا الصَّالِحَاتِ وَتَوَاصَوْا بِالْحَقِّ
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Implementation of a simple voice VoIP system for
Local Area Network (LAN).

By

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Abstract

This project is considered as the cornerstone for implementing integrated VoIP network that will be very useful to be used in large foundations and companies which has many branches in different places (these branches may be lie in a different countries).

The communication for such companies cost much money in case of using traditional telephone networks, in addition to that security is poor.

The purpose of this project is to implement VoIP network by program designed for transferring voice on pre-constructed LAN, which means that VoIP system depends on the current data networks.

This project is executed by using UDP protocol without using those protocols which were designed for VoIP especially such as SIP, H.323 and others.

The results which are obtained proved that UDP is a good to be used in case of transferring voice on LAN only.

voice was processed based on Application Programming Interfaces (APIs) functions for Windows OS.

One of the objectives of this project is that communicating between devices is performed without depending on a server; to decrease the cost and also decrease delay that will be increase in case of using server, that means the Peer-Peer mode is used.

For one device decide to communicate with other, it just need to know the IP address of that device, because the ports of both devices will be generated randomly by the program in caller side.

بناء نظام بسيط لنقل الصوت عبر بروتوكول الانترنت لشبكة محلية (LAN)

إعداد

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تجري

يعتبر هذا المشروع حجر الأساس لبناء شبكة VoIP متكاملة التي ستكون مفيدة جداً كي تستخدم في الشركات والمؤسسات الكبيرة والتي لها فروع عديدة في أماكن مختلفة (هذه الفروع قد تكون في دول مختلفة).

الاتصالات لمثل هذه الشركات تكلف أموالاً كثيرة في حالة استخدام شبكات التليفونات التقليدية، إضافة إلى ضعف أمن الاتصالات في مثل هذه الشبكات.

هذا المشروع كان الهدف منه بناء شبكة VoIP عن طريق تصميم برنامج لنقل الصوت خلال شبكة محلية (LAN) موجودة أصلاً؛ مما يعني الاعتماد على شبكات نقل البيانات الحالية. نفذ هذا المشروع باستخدام بروتوكول UDP، دون استخدام البروتوكولات المصممة خصيصاً لل VoIP مثل SIP و H.323 وغيرها.

النتائج المتحصل عليها أثبتت أن ال UDP غير جيد لنقل الصوت خلال شبكة محلية فقط.

تم الاعتماد على دوال ال API الخاصة بنظام الويندوز لمعالجة الصوت.

أحد أهداف هذا المشروع هو إجراء الاتصال وتنصيب الاتصال بين الأجهزة دون الحاجة إلى server؛ لانخفاض التكاليف وكذلك تقليل التأخير الذي يزيد في حالة استخدام ال server، أي أن النظام المستخدم هو Peer-to-Peer.

في حالة أن جهاز قرر الاتصال بأخر فإنه فقط يحتاج لمعرفة عنوان ال IP الخاص بذلك الجهاز، لأن المنافذ لكلا الجهازين يتم توليدها عشوائياً بواسطة البرنامج في جهة المتصل.

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Abbreviations

ACELP	Algebraic Code Excited Linear Predictive
APIs	Application Programming Interfaces
ARPANET	Advanced Research Projects Agency Network
CAC	Call Admission Control
codec	Compressor/Decompressor or Coder/Decoder
FTP	File Transfer Protocol
HTTP	Hypertext Transport Protocol
IANA	Internet Assigned Numbers Authority
IETF	Internet Engineering Task Force
IHL	IP Header Length
IMAP	Internet Message Access Protocol
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ISO	International Standards Organization
ITU	International Telecommunications Union.
LAN	Local Area Network
MAN	Metropolitan Area Network
Megaco	Media Gateway Control
MGCP	Media Gateway Control Protocol
MPMLQ	Multi-Pulse Maximum Likelihood Quantization
OSI	Open System Interconnection
P2P	Peer-to-Peer
PBXs	Private Branch Exchange
PCM	Pulse Code Modulation
PLC	Packet Loss Concealment
POP	Post Office Protocol version 3

PSDN	Packet Switched Data Network
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RAS	(registration, admission, and status)
RFC	Request For Comments is a formal document from the IETF
RTP	Real time Transport Protocol
RTP	Real-Time Transport Protocol
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol
SMTP	Simple Mail Transfer Protocol
TCP	Transmission Control Protocol
Telnet	Remote access protocol
TH	Transport Header
Transcoding	Conversion between different codecs
UDP	User Datagram Protocol
VLANs	Virtual LANs
VoIP	Voice over Internet Protocol
WAN	Wide Area Network